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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/702,193	10/30/2000	Ramanathan T. Jagadeesan	2705-145	9837

20575 7590 03/31/2004

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EXAMINER	
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ART UNIT	**PAPER NUMBER**
2155	4

DATE MAILED: 03/31/2004

Please find below and/or attached an Office communication concerning this application or proceeding.

PTO-90C (Rev. 10/03)

Office Action Summary	Application No.	Applicant(s)
	09/702,193	JAGADEESAN, RAMANATHAN T.
	Examiner	**Art Unit**
	Benjamin R Bruckart	2155

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *27 February 2004*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-5 and 7-44* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-5 and 7-44* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: _____.

Detailed Action

Status of Claims:

Claims 1-5, 7-44 are pending in this Office Action.

Response to Arguments

Applicant's arguments with respect to claims 1-5, 7-44 have been considered but are moot in view of the new ground(s) of rejection.

Applicant's invention as claimed:

Claim Rejections - 35 USC § 102

The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless –
>
> (b) the invention was patented or described in a printed publication in this or a foreign country or in public use or on sale in this country, more than one year prior to the date of application for patent in the United States.

Claims 1-5, 7, 8, 10-11, 17-20, 22-23, 31-34, 41-44 are rejected under 35 U.S.C. 102(b) as being anticipated by U.S. Patent No. 5,150,387 by Yoshikawa et al.

Regarding claim a transmitting device (col. 2, lines 30-36) comprising:

an input for receiving data that represents sound (col. 4, lines 44-48);

a low pass filter for selecting a first group of the data that represents a low frequency portion of the sound (col. 4, lines 58-63; col. 5, line 3);

a high pass filter for selecting a second group of the data that represents a high frequency portion of the sound (col. 4, lines 58-63, col. 5, line 4); and

a transmit buffer for transmitting to a network the first data group in a first packet and the second data group in a second packet distinct from the first packet (col. 6, lines 4-19, lines 62-col. 7, line 3).

Regarding claim 2, the device of claim 1, further comprising:

an encoder for encoding the first data group from the low pass filter into the first packet and encoding the second data group from the high pass filter into the second packet and sending the first and second packet into the transmit buffer (col. 5, lines 17-34).

Regarding claim 3, the device of claim 2, further comprising:

a switch having a first position directing the first data group from the low pass filter into the first packet, and a second position for directing the second data group from the high pass filter to the encoder then interleaving the first packet with the second packet (col. 4, lines 58-68; col. 5, lines 10-34; the QMF is the switch and filters; the encoders are tag 104; col. 6, line 4; multiplexer unit interleaves).

Regarding claim 4, the device of claim 3, further comprising:

a delay buffer for delaying the arrival to the switch of one of the first data group and the second data group (col. 4, lines 44).

Regarding claim 5,

The Yoshikawa reference teaches:

a receiving device (col. 10, lines 9-11; Figure 9) comprising:

a network interface for coupling to a network (col. 10, lines 12-20; Figure 9, tag 401); and

a processor coupled with the network interface (embedded control unit of the hardware), wherein the processor is adapted to receive a first packet and a second packet from the network (col. 10, lines 12-21),

extract a first group of data from the first packet representing a first low frequency band of a sound signal (col. 10, lines 35-49),

extract a second group of data from the second packet representing a second high frequency band of the sound signal distinct from the first band (col. 10, lines 35-49), and

combine the first data group with the second data group to construct a single data frame representing both the first high frequency band and the second low frequency band of the same sound signal (col. 23, lines 25-28).

Regarding claim 7, the device of claim 5, wherein the processor is further adapted to:

receive at least one additional packet (col. 20, lines 46-61), and

extract an additional first group of data from the additional packet representing the first low frequency band (col. 20, lines 49-56).

Regarding claim 8, the device of claim 7, wherein

the first data group is identical to the additional first data group (col. 20, lines 49-52; first frame).

Regarding claim 10, an article comprising: a storage medium (col. 4, lines 44-45; Figure 1, tag 1), said storage medium having stored thereon instructions, that, when executed by at least one device result in:

arranging data that represents sound in a plurality of frames (col. 15, lines 15-24);

dividing the data of at least one frame into a first group that represents sound within a first band of a sound bandwidth and a second group that represents sound within a second band of the sound bandwidth (col. 4, lines 58- col. 5, line 4);

encoding the first data group as a first packet (col. 5, lines 17-34);

encoding the second data group as a second packet distinct from the first packet (col. 5, lines 17-34); and

transmitting the first packet and the second packet through the network (col. 6, lines 4-19, lines 62- col. 7, line 3).

Regarding claim 11, the article of claim 10, wherein

the first band is a low-frequency band (col. 4, lines 58- col. 5, line 4), and

the second band is a high-frequency band (col. 4, lines 58- col. 5, line 4).

Regarding claim 17, an article comprising: a storage medium (col. 4, lines 44-45; Figure 1, tag 1), said storage medium having stored thereon instructions, that, when executed by at least one device, result in:

receiving a first packet and a second packet from a network (col. 10, lines 9-20);

extracting a first group of data from the first packet representing sound belonging in a first band of a sound bandwidth (col. 20, lines 46-61);

extracting a second group of data from the second packet representing sound belonging in a second band of the sound bandwidth distinct from the first band (col. 20, lines 46-61); and

combining the first data group with the second data group to construct a single frame with data representing sound in both the first band and the second band (col. 23, lines 25-28).

Regarding claim 18, an article comprising: a storage medium (col. 4, lines 44-45; Figure 1, tag 1), said storage medium having stored thereon instructions, that, when executed by at least one device, result in:

inferring a first group of data representing sound belonging in a first band of a sound bandwidth (col. 4, lines 58-63; col. 20, lines 46-61);

receiving a packet from a network (col. 10, lines 9-20);

extracting a second group of data from the packet representing sound belonging in a second band of the sound bandwidth distinct from the first band (col. 20, lines 46-61); and

combining the first data group with the second data group to construct a single frame with data representing sound in both the first band and the second band (col. 23, lines 25-28).

Regarding claim 19, the article of claim 18, wherein the instructions further result in:

receiving at least one additional packet (col. 10, lines 9-20); and

extracting an additional first group of data from the additional packet representing sound belonging in the first band (col. 20, lines 46-61),

wherein the first data group is inferred from the additional first data group (col. 10, lines 35-42; same band of signal).

Regarding claim 20, the article of claim 18, wherein

the first data group is identical to the additional data group (col. 20, lines 49-52; first frame).

Regarding claim 22, a method comprising:

arranging data that represents sound in a plurality of frames (col. 15, lines 15-24);

dividing the data of at least one frame into a first group that represents sound within a first band of a sound bandwidth and a second group that represents sound within a second band of the sound bandwidth (col. 4, lines 58- col. 5, line 4);

encoding the first data group as a first packet (col. 5, lines 17-34);

encoding the second data group as a second packet distinct from the first packet (col. 5, lines 17-34); and

transmitting the first packet and the second packet through the network (col. 6, lines 4-19, lines 62- col. 7, line 3).

Regarding claim 23, the method of claim 22, wherein

the first band is a low-frequency band (col. 4, lines 58- col. 5, line 4), and
the second band is a high-frequency band (col. 4, lines 58- col. 5, line 4).

Regarding claim 31, a method comprising:

receiving a first packet and a second packet from a network (col. 10, lines 9-20);

extracting a first group of data from the first packet representing sound belonging in a first band of a sound bandwidth (col. 20, lines 46-61);

extracting a second group of data from the second packet representing sound belonging in a second band of the sound bandwidth distinct from the first band (col. 20, lines 46-61); and

combining the first data group with the second data group to construct a single frame with data representing sound in both the first band and the second band (col. 23, lines 25-28).

Regarding claim 32, a method comprising:

inferring a first group of data representing sound belonging in a first band of a sound bandwidth (col. 20, lines 46-61);

receiving a packet from a network (col. 10, lines 9-20);

extracting a second group of data from the packet representing sound belonging in a second band of the sound bandwidth distinct from the first band (col. 20, lines 46-61); and

combining the first data group with the second data group to construct a single frame with data representing sound in both the first band and the second band (col. 23, lines 25-28).

Regarding claim 33, the method of claim 32, further comprising:

receiving at least one additional packet (col. 23, lines 25-28); and

extracting an additional first group of data from the additional packet representing sound belonging in the first band (col. 23, lines 25-28),

wherein the first data group is inferred from the additional first data group (col. 23, lines 25-28).

Regarding claim 34, the method of claim 32, wherein

the first data group is identical to the additional data group (col. 23, lines 25-28).

Regarding claim 41, a transmitting device (col. 2, lines 30-36) comprising:

input means for receiving data that represents sound (col. 4, lines 44-48);

low pass filter means for selecting a first group of the data that represents sound within a low portion of a sound bandwidth (col. 4, lines 58-63; col. 5, line 3);

high pass filter means for selecting a second group of the data that represents sound within a high portion of the sound bandwidth (col. 4, lines 58-63, col. 5, line 4); and

transmit buffer means for transmitting to a network the first data group in a first packet and the second data group in a second packet distinct from the first packet (col. 6, lines 4-19, lines 62- col. 7, line 3).

Regarding claim 42, the device of claim 41, further comprising:

encoding means for encoding the first data group and the second data group prior to transmitting it (col. 5, lines 17-34).

Regarding claim 43, the device of claim 41, further comprising:

switch means having a first position for the transmit buffer means to receive the first data group from the low pass filter, and a second position for the transmit buffer means to receive the second data group from the high pass filter (col. 4, lines 58-68; col. 5, lines 10-34; the QMF is the switch and filters; the encoders are tag 104).

Regarding claim 44, the device of claim 43, further comprising:

delay buffer means for delaying the arrival to the switch of one of the first data group and the second data group (col. 4, lines 44).

Claim Rejections - 35 USC § 103

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

Claim 9 is rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent No. 5,150,387 by Yoshikawa et al in view of U.S. Patent No. 5,467,372 by Nishitani ("Nishitani").

Claims 12, 15, 24, 27 are rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent No. 5,150,387 by Yoshikawa et al in view of U.S. Patent No. 6,389,038 by Goldberg ("Goldberg").

Claims 13-14, 16, 25, 26, 28-29 are rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent No. 5,150,387 by Yoshikawa et al in view of U.S. Patent No. 6,389,038 by Goldberg ("Goldberg") in further view of U.S. Patent No. 6,122,338 by Yamauchi ("Yamauchi").

Claim 21 is rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent No. 5,150,387 by Yoshikawa et al in view of U.S. Patent No. 6,122,338 by Yamauchi ("Yamauchi").

Claim 30 is rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent No. 5,150,387 by Yoshikawa et al in view of U.S. Patent No. 6,389,038 by Goldberg ("Goldberg") in further view of U.S. Patent No. 6,122,338 by Yamauchi ("Yamauchi") in further view of U.S. Patent No. 6,606,600 by Murgia ("Murgia").

Claims 35 and 38-40 are rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent No. 5,150,387 by Yoshikawa et al in view of U.S. Patent No. 6,606,600 by Murgia ("Murgia").

Claims 36 and 37 are rejected under 35 U.S.C. 103(a) as being unpatentable over U.S. Patent No. 5,150,387 by Yoshikawa et al in view of U.S. Patent No. 5,621,660 by Chaddha et al ("Chaddha").

Regarding claim 9,

The Yoshikawa reference teaches a system of receiving packets from a network and decoding them into a sound signal.

The Yoshikawa reference does not explicitly state the expanding of abbreviated data.

The Nishitani reference teaches, receiving abbreviated redundant data corresponding to the first data group, and expanding the received abbreviated data (Nishitani: col. 2, lines 62-67; expansion device for compressed data, compressed data is abbreviated redundant data).

The Nishitani reference further teaches that the compression and expansion reduce transmission signal bit number and increases reception in a limited time with time-sharing (Nishitani: col.2, lines 20-26).

Therefore it would have been obvious at the time of the invention to one of ordinary skill in the art to create the system of receiving packets from a network and decoding them into a sound signal as taught by Yoshikawa while employing compression and expansion as taught by Nishitani to reduce transmission signal bit number and increase reception in a limited time with time sharing (Nishitani: col.2, lines 20-26).

Regarding claim 12,

The Yoshikawa reference teaches a system of filtering sound by bandwidth and encoding the sound into packets.

The Yoshikawa reference does not explicitly state the use of data from specific frames split into two different frames.

The Goldberg reference teaches the first packet also includes data from a second frame distinct from the first frame, and the second packet also includes data from a third frame distinct from the first and second frames (Goldberg: col. 3, lines 30-40).

The Goldberg reference further teaches the use of super-packets which makes more efficient use of the bandwidth from communication protocol headers (Goldberg: col. 1, line 66- col. 2, line 2)

Therefore it would have been obvious at the time of the invention to one of ordinary skill in the art to create the system of filtering sound by bandwidth into packets as taught by Yoshikawa while employing super-packets composed of different frames of data as taught by Goldberg to make more efficient use of the bandwidth resulting from communication protocol headers through super-packets (Goldberg: col. 1, line 66- col. 2, line 2).

Regarding claims 13 and 14,

The Yoshikawa and Goldberg references teach an article of filtering sound by bandwidth into packets.

The Yoshikawa and Goldberg references do not explicitly state the expanding of abbreviated data.

The Yamauchi reference teaches wherein the instructions further result in abbreviating and transmitting redundantly the first data group through the network (Yamauchi: col. 1, lines 17-25).

The Yamauchi reference further teaches the system copes with the demand of the user and the restriction of the line speed through different bit rate compression (Yamauchi: col. 1, lines 27-31).

Therefore it would have been obvious at the time of the invention to one of ordinary skill in the art to create the article with storage of filtering sound by bandwidth into packets as taught by Yoshikawa and Goldberg while abbreviating and transmitting data as taught by Yamauchi to transmit and reproduce the audio information quickly through compression (Yamauchi: col. 1, lines 27-31).

Claim 14 is rejected under the same rationale given above. In the rejections set fourth, the examiner will address the additional limitations and point to the relevant teachings of Tan and Yamauchi.

Regarding claim 14, the article of claim 13, wherein the instructions further result in: abbreviating includes down sampling the first data group (Yamauchi: col. 1, lines 43-47).

Regarding claim 15,

The Yoshikawa reference teaches an article comprising: a storage medium (col. 4, lines 44-45; Figure 1, tag 1), said storage medium having stored thereon instructions, that, when executed by at least on device result in:

receiving three sequential frames of data that represent sound (col. 15, lines 15-23);

dividing the data of each of the three frames into a first group that represents sound within a low band of a sound bandwidth and a second group that represents sound within a high band of the sound bandwidth (col. 4, lines 58- col. 5, line 4);

transmitting the first and second packets through the network (col. 6, lines 4-19, lines 62- col. 7, line 3).

The Goldberg reference teaches encoding the first data group of the first frame and the second data group of the second frame as a first packet (Goldberg: col. 3, lines 30-40) and encoding the first data group of the second frame and the second data group of the third frame as a second packet (Goldberg: col. 3, lines 30-40).

The Goldberg reference further teaches the use of super-packets makes more efficient use of the bandwidth from communication protocol headers (Goldberg: col. 1, line 66- col. 2, line 2)

Therefore it would have been obvious at the time of the invention to one of ordinary skill in the art to create the system of receiving frames of sound and dividing them by frequency as

taught by Yoshikawa while employing super-packets composed of different frames of data as taught by Goldberg to make more efficient use of the bandwidth resulting from communication protocol headers (Goldberg: col. 1, line 66- col. 2, line 2).

Regarding claim 16,

The Yoshikawa and Goldberg references teach an article that filters sound by bandwidth into packets.

The Yoshikawa and Goldberg references do not explicitly state the expanding of abbreviated data.

The Yamauchi reference teaches wherein the instructions further result in abbreviating and transmitting redundantly the first data group through the network (Yamauchi: col. 1, lines 17-25).

The Yamauchi reference further teaches the system copes with the demand of the user and the restriction of the line speed through different bit rate compression (Yamauchi: col. 1, lines 27-31).

Therefore it would have been obvious at the time of the invention to one of ordinary skill in the art to create the article with storage of filtering sound by bandwidth into packets as taught by Yoshikawa and Goldberg while abbreviating and transmitting data as taught by Yamauchi to transmit and reproduce the audio information quickly through compression (Yamauchi: col. 1, lines 27-31).

Regarding claim 21,

The Yoshikawa reference teaches a system of filtering sound by bandwidth and encoding the sound into packets.

The Yoshikawa reference does not explicitly state abbreviating and expanding redundant data.

The Yamauchi reference teaches wherein the instructions further result in abbreviating and expanding redundant data from the first group through the network (Yamauchi: col. 1, lines 17-25).

The Yamauchi reference further teaches the system copes with the demand of the user and the restriction of the line speed through different bit rate compression (Yamauchi: col. 1, lines 27-31).

Therefore it would have been obvious at the time of the invention to one of ordinary skill in the art to create the article with storage of filtering sound by bandwidth into packets as taught by Yoshikawa and Goldberg while abbreviating and transmitting data as taught by Yamauchi to transmit and reproduce the audio information quickly through compression (Yamauchi: col. 1, lines 27-31).

Regarding claim 24,

The Yoshikawa reference teaches a system of filtering sound by bandwidth and encoding the sound into packets.

The Yoshikawa reference does not explicitly state a packet also includes data from a second frame distinct from the first frame and a second packet also includes data from a third frame distinct from the first and second frames.

The Goldberg reference teaches encoding the first packet also includes data from a second frame distinct from the first frame (Goldberg: col. 3, lines 30-40), and the second packet also includes data from a third frame distinct from the first and second frames (Goldberg: col. 3, lines 30-40).

The Goldberg reference further teaches the use of super-packets makes more efficient use of the bandwidth from communication protocol headers (Goldberg: col. 1, line 66- col. 2, line 2)

Therefore it would have been obvious at the time of the invention to one of ordinary skill in the art to create the system of receiving frames of sound and dividing them by frequency as taught by Yoshikawa while employing super-packets composed of different frames of data as taught by Goldberg to make more efficient use of the bandwidth resulting from communication protocol headers (Goldberg: col. 1, line 66- col. 2, line 2).

Regarding claim 25,

The Yoshikawa and Goldberg references teach a system of filtering sound by bandwidth and encoding frames of data into different packets.

The Yoshikawa and Goldberg references do not explicitly state abbreviating and transmitting redundantly the first data group through the network.

The Yamauchi reference teaches abbreviating and transmitting redundantly the first data group through the network (Yamauchi: col. 1, lines 17-25).

The Yamauchi reference further teaches the system copes with the demand of the user and the restriction of the line speed through different bit rate compression (Yamauchi: col. 1, lines 27-31).

Therefore it would have been obvious at the time of the invention to one of ordinary skill in the art to create the article with storage of filtering sound by bandwidth into packets as taught by Yoshikawa and Goldberg while abbreviating and transmitting data as taught by Yamauchi to transmit and reproduce the audio information quickly through compression (Yamauchi: col. 1, lines 27-31).

Claim 26 is rejected under the same rationale given above. In the rejection set fourth, the examiner will address the additional limitations and point to the relative teaches of Yoshikawa, Goldberg and Yamauchi.

Regarding claim 26, the method of claim 25, wherein

abbreviating includes down-sampling the first data group (Yamauchi: col. 1, lines 43-47).

Regarding claim 27,

The Yoshikawa reference teaches a method comprising:

receiving three sequential frames of data that represent sound (col. 15, lines 15-23);

dividing the data of each of the three frames into a first group that represents sound within a low band of a sound bandwidth and a second group that represents sound within a high band of the sound bandwidth (col. 4, lines 58- col. 5, line 4);

transmitting the first and second packets through the network (col. 6, lines 4-19, lines 62- col. 7, line 3).

The Yoshikawa does not explicitly state encoding the first data group of the first frame and the second data group of the second frame as a first packet and encoding the first data group of the second frame and the second data group of the third frame as a second packet.

The Goldberg reference teaches encoding the first data group of the first frame and the second data group of the second frame as a first packet (Goldberg: col. 3, lines 30-40) and encoding the first data group of the second frame and the second data group of the third frame as a second packet (Goldberg: col. 3, lines 30-40).

The Goldberg reference further teaches the use of super-packets makes more efficient use of the bandwidth from communication protocol headers (Goldberg: col. 1, line 66- col. 2, line 2)

Therefore it would have been obvious at the time of the invention to one of ordinary skill in the art to create the system of receiving frames of sound and dividing them by frequency as taught by Yoshikawa while employing super-packets composed of different frames of data as taught by Goldberg to make more efficient use of the bandwidth resulting from communication protocol headers (Goldberg: col. 1, line 66- col. 2, line 2).

Regarding claim 28,

The Yoshikawa and Goldberg references teach a system of filtering sound by bandwidth and encoding frames of data into different packets.

The Yoshikawa and Goldberg references do not explicitly state abbreviating and transmitting redundantly the first data group through the network.

The Yamauchi reference teaches abbreviating and transmitting redundantly the first data group through the network (Yamauchi: col. 1, lines 17-25).

The Yamauchi reference further teaches the system copes with the demand of the user and the restriction of the line speed through different bit rate compression (Yamauchi: col. 1, lines 27-31).

Therefore it would have been obvious at the time of the invention to one of ordinary skill in the art to create the article with storage of filtering sound by bandwidth into packets as taught

by Yoshikawa and Goldberg while abbreviating and transmitting data as taught by Yamauchi to transmit and reproduce the audio information quickly through compression (Yamauchi: col. 1, lines 27-31).

Claim 29 is rejected under the same rationale given above. In the rejection set fourth, the examiner will address the additional limitations and point to the relative teaches of Yoshikawa, Goldberg and Yamauchi.

Regarding claim 29, the method of claim 28, wherein

abbreviating includes down-sampling (Yamauchi: col. 1, lines 43-47).

Regarding claim 30,

The Yoshikawa, Goldberg and Yamauchi references teach a system of filtering audio data and then combining the data into super-packets for transmission across the network with abbreviated data like compression and header information.

The Yoshikawa, Goldberg, and Yamauchi references do not specifically state the use of complementary band information synthesis shift.

The Murgia reference teaches, abbreviating includes determining a complementary band information synthesis shift between one of the first data group and one of the second data group (Murgia: col. 15, lines 37-51; the decoder reconstructs the components according to the two receiver bands).

The Murgia reference further teaches it allows for variable bit rate codecs for greater precision to be achieved in the permitted frequency variations (Murgia: col. 2, lines 64-67) and a reduction in bit rate without an appreciable loss of quality (Murgia: col. 4, lines 39-48).

Therefore it would have been obvious at the time of the invention to one of ordinary skill in the art to create the system transmitting data through filters and encoders onto a network into super-packets as abbreviated data as taught by Tan, Goldberg, and Yamauchi while employing a complementary band information synthesis shift as taught by Murgia to allow for variable bit rate codecs for greater precision to be achieved in the permitted frequency variations (Murgia:

col. 2, lines 64-67) and a reduction in bit rate without an appreciable loss of quality (Murgia: col. 4, lines 39-48).

Regarding claim 38,

The Yoshikawa reference teaches a system of transmitting data through filters and encoding them before sending them through a network.

The Yoshikawa reference does not explicitly state the use of a complementary band information synthesis shift.

The Murgia reference teaches using a complementary band information synthesis shift to infer data in the first band from data in the second band (Murgia: col. 15, lines 37-51; the decoder reconstructs the components according to the two receiver bands).

The Murgia reference further teaches it allows for variable bit rate codecs for greater precision to be achieved in the permitted frequency variations (Murgia: col. 2, lines 64-67) and a reduction in bit rate without an appreciable loss of quality (Murgia: col. 4, lines 39-48).

Therefore it would have been obvious at the time of the invention to one of ordinary skill in the art to create the system transmitting data through filters and encoders onto a network as taught by Tan while employing a complementary band information synthesis shift as taught by Murgia to allow for variable bit rate codecs for greater precision to be achieved in the permitted frequency variations (Murgia: col. 2, lines 64-67) and a reduction in bit rate without an appreciable loss of quality (Murgia: col. 4, lines 39-48).

Claims 35, 39 and 40 are rejected under the same rationale given above. In the rejections set fourth, the examiner will address the additional limitations and point to the relevant teachings of Tan and Murgia et al.

Regarding claim 35, the method of claim 32, wherein the first data group is determined from a weighted average that includes the additional data group (Murgia: col. 11, lines 14-24).

Regarding claim 39, the method of claim 38, further comprising: receiving and decoding the complementary band information synthesis shift (Murgia: col. 15, lines 37-51; the decoder reconstructs the components according to the two receiver bands).

Regarding claim 40, the method of claim 38, further comprising:

determining the complementary band information synthesis shift from at least one other received first data group and at least one received second data group (Murgia: col. 15, lines 37-51; the decoder reconstructs the components according to the two receiver bands).

Regarding claim 36,

The Yoshikawa reference teaches a system of transmitting data through filters and encoding them before sending them through a network.

The Yoshikawa reference does not explicitly state the use of an abbreviated data.

The Chaddha reference teaches wherein inferring is performed by: receiving abbreviated redundant data corresponding to the first data group (Chaddha: col. 3, lines 37-48; receiving end, decoders...); and expanding the received abbreviated data (Chaddha: col. 3, lines 37-48; decompressing).

The Chaddha reference further teaches this system accommodates lower bandwidth links and congestion and permits the encoder to operate independently of decoder capability (Chaddha: col. 2, lines 52-55)

Therefore it would have been obvious at the time of the invention to one of ordinary skill in the art to create the system transmitting data through filters and encoders onto a network as taught by Tan while employing receiving and expanding abbreviated data as taught by Chaddha to accommodate lower bandwidth links and congestion and permits the encoder to operate independently of decoder capability (Chaddha: col. 2, lines 52-55).

Claim 37 is rejected under the same rationale given above. In the rejections set fourth, the examiner will address the additional limitations and point to the relevant teachings of Tan and Chaddha et al.

Regarding claim 37, the method of claim 36, wherein expanding includes up-sampling the abbreviated data (Chaddha: col. 3, lines 37-48; decompressing, up-sampling).

Prior Art

The prior art made of record and not relied upon is considered pertinent to applicant's disclosure:

U. S. Patent No. 5,231,669 issued to Galand et al. Galand teaches a low bit rate voice encoding device.

PCT Application WO 98/52187 issued to Tucker et al. Tucker teaches the main invention also with filtering frames.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Benjamin R Bruckart whose telephone number is (703) 305-0324. The examiner can normally be reached on 8:00-5:30 PM with every other Friday off.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Hosain Alam can be reached on (703) 308-6662. The fax phone numbers for the organization where this application or proceeding is assigned are (703) 872-9306 for regular communications and After Final communications.

Any inquiry of a general nature or relating to the status of this application or proceeding should be directed to the receptionist whose telephone number is (703) 305-0324.

Benjamin R Bruckart
Examiner
Art Unit 2155
brb
March 24, 2004

BRB

HOSAIN ALAM
SUPERVISORY PATENT EXAMINER